CLAIMS

1. An end-to-end estimation of the bandwidth available in a client-server connection established over a packet switching network, comprising:

a routine to compute samples of available bandwidth by taking into accounts the flow of packets received by the client, if the routine is implemented at the receiver side, or by taking into accounts acknowledgment packets received by the sender, if the routine is implemented at the sender side;

a routine that implements a discrete time low-pass filter to obtain a filtered value of the samples of available bandwidth.

2. The end-to-end bandwidth estimation according to claim **1**, wherein a sample of available bandwidth b_j at time t_j is computed as:

$$b_j = \frac{d_j}{t_j - t_{j-1}}$$

where d_j is the amount of data that have been received at the receiver or acknowledged at the sender in the interval $t_j - t_{j-1}$, t_{j-1} is the time when the previous ACK was received by the sender or the time when the previous packet was received by the receiver, and t_j is the time when the current ACK is received by the sender or when the current packet is received by the receiver.

3. The end-to-end bandwidth estimation according to claim **1**, wherein the routine implements a discrete time low-pass filter with time-varying coefficients.

4. The end-to-end bandwidth estimation according to claim **1**, wherein the routine implements the discrete-time low-pass filter with time-varying coefficients:

$$\hat{b}_j = \frac{2\tau_f - \Delta_j}{2\tau_f + \Delta_j}\hat{b}_{j-1} + \Delta_j \frac{b_j + b_{j-1}}{2\tau_f + \Delta_j}$$

where $\hat{b}_j$ is the filtered measurement of the available bandwidth at time $t = t_j$, $\hat{b}_{j-1}$ is the filtered measurement of the available bandwidth at time t_{j-1}, $\Delta_j = t_j - t_{j-1}$, $1/\tau_f$ is the cut-off frequency of the filter, b_j is the sample of the available bandwidth at time t_j, and b_{j-1} is the sample of the available bandwidth at time t_{j-1}. If a time τ_f/m $(m \geq 2)$ has

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elapsed since the last received ACK or packet without receiving any new ACK or packet, then the filter assumes the reception of a *virtual* sample $b_i=0$.

5. Method for adapting the amount of data for unit of time, i.e. the rate, sent by the server to the client over a packet network, comprising an end-to-end bandwidth estimation according to claim 1.

6. Method for adapting the amount of data for unit of time sent by the server to the client according to claim 5, wherein the low pass filter is a low pass filter according to claim 3 or 4.

7. Method for adaptively setting congestion window and slow start threshold in the TCP/IP protocol, comprising an end-to-end bandwidth estimation according to claim 1.

8. Method for adaptively setting congestion window and slow start threshold in the TCP/IP protocol according to claim 7, wherein the low pass filter is a low pass filter according to claim 3 or 4.

9. Method for adaptively selecting the quality of coding, or the numbers of layers to be transmitted in a layered coding of an audio/video source using the TCP protocol, or the UDP protocol or the RTP protocol, comprising an end-to-end bandwidth estimation according to claim 1.

10. Method for adaptively selecting the quality of coding, or the numbers of layers to be transmitted in a layered coding of an audio/video source using the TCP protocol, or the UDP protocol or the RTP protocol according to claim 9, wherein the low pass filter is a low pass filter according to claim 3 or 4.

11. Method for adaptively selecting the quality of coding, or the numbers of layers to be transmitted in a layered coding of an audio/video source according to claim 9 or 10, comprising:

increasing step by step the quality of coding, or the numbers of layers to be transmitted in a layered coding of an audio/video source until congestion is experienced by means of control packets;

setting the quality of coding or select the numbers of layers to be transmitted after that a congestion episode is signaled by means of control packets in according with the bandwidth estimation according to any of claims 1 to 4;

increasing again step by step the quality of coding or the numbers of layers to be transmitted in a layered coding to probe for extra available bandwidth.

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12. Method for setting the Advertised Window of TCP equal to the minimum of the Advertised Window and the bandwidth estimate times the minimum round trip time, wherein the samples are computed according to claim **1** or **2**, and the bandwidth estimate is computed according to any of claim **1, 3** or **4**.